AMENDMENTS TO THE CLAIMS

Claims 1-33 are pending in the Application and all were allowed in the Notice of Allowance mailed September 7, 2006. No claims are amended in this paper.

Listing of Claims:

Claim 1. (Original) An audio conferencing system comprising:

at least one loudspeaker for converting a first electrical signal into sound;

a plurality of conference stations in spaced relation, each conference station comprising a directional microphone for converting sound into a directional microphone signal, the directional microphone signals collectively forming a plurality of directional microphone signals; and

a signal processor for modifying at least one of the plurality of directional microphone signals and a receive signal, the signal processor producing at least one of a transmit signal and the first electrical signal.

Claim 2. (Original) The audio conferencing system of claim 1 wherein the modifying comprises an algorithm to perform acoustic echo cancellation.

Claim 3. (Original) The audio conferencing system of claim 1 wherein the modifying comprises an adaptive beamforming technique.

Claim 4. (Original) The audio conferencing system of claim 3 wherein the adaptive beamforming technique comprises at least one of a normalized least mean squares algorithm and a recursive least squares algorithm.

Claim 5. (Original) The audio conferencing system of claim 1 wherein the modifying combines the plurality of directional microphone signals in order to selectively attenuate or amplify a sound source.

Claim 6. (Original) The audio conferencing system of claim 1 wherein the modifying selects for separate processing at least two groups of directional microphone signals from the plurality of directional microphone signals.

Claim 7. (Original) The audio conferencing system of claim 6 wherein the modifying of each of the at least two groups uses an adaptive beamforming technique.

Claim 8. (Original) The audio conferencing system of claim 1 further comprising at least one omni-directional microphone for converting a sound field into an omni-directional microphone signal.

Claim 9. (Original) The audio conferencing system of claim 8 wherein the modifying comprises combining at least one of the plurality of directional microphone signals and the at least one omni-directional microphone signal, based upon at least one room condition.

Claim 10. (Original) The audio conferencing system of claim 9 wherein the at least one room condition comprises at least one of background noise, a level of acoustic echo, and the detection of side conversations.

Claim 11. (Original) The audio conferencing system of claim 1 wherein each of the conference stations comprises a transducer for producing an acoustic test signal.

Claim 12. (Original) The audio conferencing system of claim 1 wherein the signal processor uses a test signal to determine at least one of microphone and room acoustic characteristics.

Claim 13. (Original) The audio conferencing system of claim 1 wherein the contribution to the transmit signal of a selected sound source relative to other sound sources may be increased or decreased from a location remote from the audio conferencing system.

Claim 14. (Original) The audio conferencing system of claim 1 further comprising an interface compatible with a communication network, the interface coupling the transmit signal to the communication network, and the communication network to the receive signal.

Claim 15. (Original) The audio conferencing system of claim 14 wherein the communication network is a packet network.

Claim 16. (Original) The audio conferencing system of claim 1 further comprising a manual input device used for at least one of controlling calls and entering system parameters.

Claim 17. (Original) The audio conferencing system of claim 1 wherein the signal processor is a digital signal processor.

Claim 18. (Original) A method of operating an audio conferencing system comprising:

receiving a first electrical signal;

transducing each of a plurality of sound fields into a microphone signal, the microphone signals collectively forming a plurality of microphone signals;

processing at least one of the plurality of microphone signals and the first electrical signal to produce a second electrical signal; and

transmitting the second electrical signal.

Claim 19. (Original) The method of claim 18 wherein the processing comprises an algorithm to perform acoustic echo cancellation.

Claim 20. (Original) The method of claim 18 wherein the processing comprises an adaptive beamforming technique.

Claim 21. (Original) The method of claim 20 wherein the adaptive beamforming technique comprises at least one of a normalized least mean squares algorithm and a recursive least squares algorithm.

Claim 22. (Original) The method of claim 18 wherein the processing comprises selecting at least two groups of microphone signals from the plurality of microphone signals.

Claim 23. (Original) The method of claim 22 wherein each of the at least two groups of microphone signals is used in a separate adaptive beamforming arrangement.

Claim 24. (Original) The method of claim 18 wherein the processing uses at least one parameter representative of at least one of a microphone acoustic characteristic, a transmission delay, and an acoustic characteristic of a room.

Claim 25. (Original) The method of claim 18 wherein the processing may be modified remotely during operation.

Claim 26. (Original) The method of claim 18 wherein the processing is performed using a digital signal processor.

Claim 27. (Original) The method of claim 18 wherein at least one of the first electrical signal and the second electrical signal are a digital signal.

Claim 28. (Original) The method of claim 27 wherein at least one of the first electrical signal and the second electrical signal are compliant with a packet protocol.

Claim 29. (Original) The method of claim 18 further comprising:

generating a first electrical test signal;

converting the first electrical test signal to an acoustic test signal at a first location;

sampling the acoustic test signal at a second location;

transforming the sampled acoustic test signal into a second electrical test signal; and

deriving at least one of a microphone acoustic characteristic, a transmission delay, and an acoustic characteristic of a room using the second electrical test signal.

Claim 30. (Original) A method of operating an audio conferencing system comprising:

receiving a plurality of microphone signals;

selecting at least two groups of microphone signals from the plurality of microphone signals;

processing each of the at least two groups of microphone signals using an adaptive beamforming technique, the processing producing an output signal for each of the at least two groups of microphone signals; and

combining the output signals.

Claim 31. (Original) The method of claim 30 further comprising:

performing acoustic echo cancellation on at least a portion of the plurality of microphone signals.

Claim 32. (Original) The method of claim 30 wherein the selecting is based upon at least one of an amplitude of a microphone signal, a propagation delay, and an input from a user.

Claim 33. (Original) The method of claim 30 wherein the adaptive beamforming technique comprises at least one of a normalized least mean squares algorithm and a recursive least squares algorithm.